ELECTRONIC COMMUNICATIONS SYSTEM AND METHOD

This application claims the benefit of the earlier filing date of provisional application number 60/270,261 filed on February 20, 2001.

FIELD OF THE INVENTION

The present invention relates to systems and methods for electronic communications and, more particularly, relates to systems and methods for transmitting and storing voice, facsimile or other data across both circuit-switched and packet-switched communications networks.

BACKGROUND OF THE INVENTION

The conventional telecommunications landscape is centered around the Public Switched Telephone Network (PSTN). The PSTN is a series of local communications networks interconnected to each other directly and through a wide-ranging long distance telephone network. Generally, a caller places a call from a telephone or other electronic communications equipment (telephonic device), and the call data travels over copper twisted pair or other low bandwidth medium to a local telephone switch serving a local community of telephone users (the "local loop") where it is received by a telephone line card.

If the caller is attempting to reach another person who is part of the same local telephone network, the local switch usually will route the call directly back to the callee and ring his or her telephone, completing the telephonic loop with the callee's copper twisted pair. If, on the other hand, the caller is attempting to contact a person on another local network (or even in another country), the caller's local switch will typically transfer the call to the long distance telephone system, which interconnects the various local switches using large bandwidth (high volume) communications channels such as fiberoptic cable, and route the call to the callee's local telephone switch. The callee's local switch may then take the call off of the long distance network and route the call to the correct callee telephone or other device. The routing of the call is generally guided by the telephone number (country code, area code, local exchange, and direct number) according to the Signaling System 7 (SS7) or some other transmission protocol.

The PSTN is generally a circuit-switched communications network. As such, the entire bandwidth of the system is divided into smaller segments of bandwidth called "channels" upon which calls are transmitted. As calls are received by the system, each call is allotted a specific channel and utilizes the full bandwidth within that channel for the duration of that call, even during times when the parties to the call are not speaking. New calls are assigned to additional channels as the calls are received until the entire bandwidth of the system is exhausted. Thereafter, the next caller will receive a "circuit busy" signal indicating

that the system is full. Therefore, as new callers utilize the system, these new callers may receive a "circuit busy" signal, but additional users generally will not degrade the fidelity of any of the telephone calls.

This circuit-switched system is somewhat inefficient because a caller may not utilize the entire bandwidth of his or her assigned channel throughout the conversation. Specifically, the same channel bandwidth is utilized whether or not the parties to the conversation are currently speaking. Valuable bandwidth, that could be used for other calls, is wasted.

With the creation of the World Wide Web (WWW) and web-specific applications, the Internet has exploded into a global network of research, business, and personal users. The Internet is generally a collection of smaller data networks utilizing the Internet Protocol, or "IP," transmission format. Unlike the circuit-switched architecture of the PSTN, IP networks are data packet networks (packet-switched) networks. The computers on the network are interconnected by persistent connections, the bandwidth of which is shared by all active users. Specifically, as opposed to assigning a predefined amount of the network's bandwidth (a channel) to each user, all of the users on the system send packets of information across the medium as needed. When a user is not sending or receiving data, that particular user does not waste any bandwidth. As the network gets busier, each user will remain connected but may experience performance degradation (due to increased data

traffic). IP has become the de facto standard for data networking, but new transmission protocols are always being defined.

The rise of IP-based data communications and the recent global market liberalization have changed the traditional telecommunications landscape. IP presents the ability to converge existing telephony systems and data networks into a single communication medium. IP may become the future global network for voice, fax, and data communications by providing a seamless, worldwide connectivity solution between voice and data networks.

Traditional telephonic communications systems are becoming more accessible to innovation. In the United States, Europe and Asia, including Korea, Japan, Hong Kong and Singapore, governments are deregulating aspects of the telecommunications industry to allow more people to utilize these vast communications networks. Deregulation provides broader, cheaper access to the PSTN, encouraging communications innovation.

Additionally, the IP world itself is growing at an incredible pace. More users are getting “wired” everyday, and Internet users are utilizing the medium in growing ways. Also, wireless IP applications, such as Personal Digital Assistants (PDAs) and IP-enabled pagers, are being utilized by a growing number of people. Soon, analysts predict, the IP network will be larger than the traditional telecommunications networks. The present invention attempts to utilize advantages of both of

these networking technologies to develop an integrated data communications system.

Although the PSTN and IP networks are fundamentally different in terms of routing and performance, it is possible for the networks to be interconnected, exchanging voice and data traffic. Voice over Internet Protocol (VoIP or V/IP) is an example of telephonic technology that utilizes both network types. In VoIP, a voice signal is converted into compressed data packets that are sent by IP to a receiving destination where the packets are decompressed and reassembled. The service providers on both ends of the call communicate with their respective IP telephony switches using ISDN, T1 and/or E1 connections, which include both the voice and the signaling information. On the IP networks, users send and receive voice data using an IP telephony terminal, which is typically a multimedia IP device equipped with telephony software.

Although certain VoIP systems allow voice calls to be placed between two multimedia devices, or even from a multimedia device to a traditional telephone, the use and flexibility of these systems is limited. For example, because of the lack of fully defined standards, prior VoIP products generally require that network nodes (which connect the PSTN to the IP network) from the same vendor exist on both sides of the IP network. Additionally, these prior systems do not allow a single subscriber identifier (telephone number) to be used to call a person on their IP device (PC) as well as on their landline telephone or other

communications device. Prior VoIP systems may not offer voice mail and facsimile features.

These various limitations to the current implementation of VoIP communications are preferably improved in relation to the prior art through the use of the current invention. These and other objects and advantages of the present invention will become readily apparent to persons skilled in the art from the following description of a particularly preferred embodiment.

SUMMARY OF THE INVENTION

In accordance with the present invention, there is provided a system and method for electronic communications; more specifically, a system and method for transmitting voice and facsimile electronic data across both circuit-switched and packet-switched networks is provided. The system preferably includes a gatekeeper server capable of communicating with a system subscriber across a packet-switched network and a network node capable of communicatively interconnecting a circuit-switched network, such as the PSTN, and a packet-switched network, such as the Internet, which utilizes the Internet Protocol for transmission. The system may also include client software running on a multimedia IP device that enables a subscriber to interact with the gatekeeper.

A subscriber to the system is preferably identified by a single subscriber identifier, which is preferably a telephone number that corresponds to the dialing practices of the local PSTN telephone exchange to which the subscriber is connected. By directing a communication to this subscriber identifier or to a traditional PSTN telephone number, the system allows voice calls, facsimiles, and other electronic messages to be sent to and received from both subscribers and non-subscribers on the same or other communications networks. This system preferably enables four types of seamless communications: (1) IP device to IP device; (2) IP device to telephonic device; (3) telephonic device to IP device; and (4) telephonic device to telephonic device.

If a subscriber is not available to receive an electronic communication at his primary telephone number or IP address, the system (via the network node) may be capable of re-directing the communication to additional telephone numbers or IP addresses across either the circuit-switched or packet-switched networks, according to the subscriber's preferences. For example, if a call is placed to a user's subscriber identifier (subscriber number) and that user is not logged into the packet-switched network (*e.g.*, the Internet), the network node may switch the call out over the PSTN and ring the same user on his landline phone or mobile telephone. This switching is enabled because the system determines a user's dynamic IP address when the user logs into the system and associates this IP address with the subscriber identifier in a database. Preferably, this multi-network switching is seamless to the

caller, whether the caller places the initial call from a regular telephone or an IP device.

The IP communications system may also include media servers, databases, notification servers, and other hardware and software to impart functionality to the communications system. The system may provide for non-real-time voice messaging (*e.g.*, voice mail) in the event an immediate direct connection is not available or desired. The system network nodes may also recognize incoming fax messages and store/replay these messages according to user preferences. The system components are preferably designed to be scalable, redundant and adapted to be replaced and upgraded while the system is running.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention and its presently preferred embodiments will be better understood by reference to the detailed disclosure hereinbelow and to the accompanying drawings, wherein:

Figure 1 is a high level block diagram of one embodiment of the present invention;

Figure 2 is a system architecture block diagram of one embodiment of the present invention;

Figure 3 is a functional diagram of one embodiment of a network node;

Figure 4 is a functional diagram of one embodiment of the present invention;

Figure 5 is a detailed description of one embodiment of a network node;

Figure 6 is a functional diagram of the hardware and software on the multimedia software client running on an IP device; and

Figure 7 shows a representation of one user interface of the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

IP telephony, the technology that enables telephony-based applications and devices to use existing data networks via the Internet Protocol, is creating dramatic changes in the telecommunications industry. IP telephony represents the convergence of circuit-switched networks, such as the traditional Public Switched Telephone Network (PSTN) and leased T1 and E1 lines, with packet-switched networks, such as the Internet or local intranets. The IP communications system of the present invention generally utilizes a network node, a gatekeeper and a multimedia software client running on one or more IP devices to enable interoperability of these two disparate network types and to support

communication connections between telephones, fax machines, other telephonic devices and multimedia IP devices. The present system generally facilitates four communications schemes: telephonic device to telephonic device; telephonic device to IP device; IP device to telephonic device, and IP device to IP device.

The present invention broadly contemplates, in at least one preferred embodiment, a system and method for sending, receiving, and/or storing electronic messages over both a circuit-switched network and a packet-switched network. To aid in the comprehension of the present invention, the description of the preferred embodiments is parsed into three discrete sections. Initially, a general system overview will provide a high level description of certain preferred features and elements of the invention. Next, a more detailed discussion of the hardware and software system architecture will provide a more finite understanding of the invention. Finally, a series of examples of use of at least one preferred embodiment will be presented to tie all of the described features together. It should also be noted that the physical structure and certain programming techniques utilized in the network nodes, gatekeeper, client software and other devices of the present invention are based, in part, on existing VoIP technology well-known in the art.

1. System Overview

The IP communications system of the present invention generally allows a subscriber to the system to exchange electronic

communications with other people across disparate communications networks. This functionality may allow both subscribers to the system and non-subscribers to communicate with each other in a variety of ways according to **FIG. 1**. As shown in **FIG. 1**, the IP communications system **100** uses one or more network nodes **114**, **116** to interconnect the Public-Switched Telephone Network **102** with an IP network **104** such as the Internet. When placing or receiving calls from an IP device (*e.g.*, a multimedia personal computer **106**), the subscriber utilizes client software that communicates with a gatekeeper server **108** which is the user's entry point into the IP communications system **100**. The gatekeeper **108** can interact with the network node **114**.

In one example, a caller who is a subscriber A to the system **100** may place a "telephone" call to a "callee" subscriber B of the system **100** via the caller's multimedia IP device **106**. A gatekeeper server (gatekeeper) **108**, assigned to caller A, in the system **100** preferably analyzes the placed call and determines whether or not callee B is a subscriber to the IP communications system **100**.

Because, in this case, the callee B is a subscriber, the gatekeeper **108** may poll a system database (not shown) to determine the dynamic IP address of the callee subscriber B. If the callee B is logged onto the Internet, the callee's gatekeeper **110** may poll the callee's IP device **112** and authenticate the callee B as the intended second party to the call. After authentication, the gatekeeper **108** preferably signals to the caller's client **106** and the callee's client **112** to set up a telephone call

(utilizing H.323 or some other call protocol) between the two subscribers. This call will preferably take place exclusively over the IP network **104** without any data traveling out over the PSTN **102**. This is an IP device **106** to IP device **112** call.

If the callee subscriber B was not logged in or was not available at his or her IP device **112**, the IP communications system **100** may have the capability to redirect the incoming call to the callee's mobile telephone, landline phone, pager, or some other communications device **120** (preferably over the PSTN **102**) according to the callee's preconfigured preferences. In this way, the callee's single "subscriber identifier" (which may be a telephone number) allows calls to "follow" the callee B to other communications devices **120**, thereby increasing the likelihood that the callee B receives the "telephone" call. If the callee's mobile phone, pager and/or other devices **120** are connected to the PSTN **102** (which is most likely the case), the callee's network node **116** will preferably convert and redirect the call data from the packet-switched IP network **104** to the circuit-switched PSTN **102**. As the call proceeds, the voice data must be continually converted back and forth among these two disparate networks **102**, **104**, perhaps utilizing encryption technology. This is an IP device **106** to telephonic device **120** call.

If the callee B still cannot be found, or if the callee's preferences so state, the IP communications system **100** may also have voice mail or other messaging capabilities that prompt the caller A to record a spoken message for the callee B's later retrieval. Again, the

callee B may be notified of this waiting message in a variety of ways. Thereafter, the caller B may retrieve his voice mail message using an IP device or from a regular PSTN-based telephone.

In another scenario, the subscriber caller A may attempt to communicate with a non-subscriber C by placing a call from his or her IP device **106** to the callee's telephone **122** (via the callee's PSTN telephone number). When the subscriber caller A places the call on his or her multimedia IP device **106**, the gatekeeper **108** will again preferably check a registration database to see if the intended callee C is a subscriber. In this case, the gatekeeper **108** will confirm that the callee C is not a subscriber, and the gatekeeper **108** will signal to the network node **114** to place the call on the PSTN **102** rather than internally on the IP network **104**. The network node **114** will preferably transfer the call out to the PSTN **102** and the non-subscriber callee's telephone **122** (or other device) will ring according to conventional telephone practices. From the caller A's point of view, the network node's **114** routing of the call should be substantially seamless, and the caller A should not be able to tell whether the call was transferred or not. This is an example of an IP device **106** to telephonic device **122** call.

Because the single "subscriber identifier" (which is preferably a telephone number) is a convenient way to locate a person, and further because the network node may route the call to "follow" the subscriber when he or she is away from his or her IP device, both subscribers and non-subscribers may call subscribers from regular telephones. For

example, a non-subscriber caller C may call a subscriber B via his or her subscriber number. The subscriber number preferably corresponds to the local dialing practices of the PSTN **102** at the subscriber's geographic location (*e.g.*, country code, area code, local exchange). Therefore, the PSTN **102** will route the dialed call from the caller's local loop **124** and across the long distance lines **126** to the local telephone network **128** that corresponds to the subscriber number. The IP communications system network node or nodes **116** attached to that local telephone loop **128** will preferably recognize the subscriber telephone number as belonging to a subscriber and transfer the call from the PSTN **102** to the IP-switched network **104**. The gatekeeper **110** will then preferably determine the dynamic IP address of the callee subscriber B (from a database) and ring his IP device **112**. If the callee B is connected to the IP network **104** and is properly authenticated, the network node **116** preferably sets up the telephone call from the non-subscriber C to the subscriber B. This is an example of a telephonic device **122** to IP device **112** call.

Again, if the callee subscriber B is not available to receive the Internet call, the call may be redirected by the network node **116** out of the system (over the PSTN **102**) to locate the subscriber B on a different communications device **120**. In this case, the call would now continue over only the PSTN **102**. The call would travel like a regular telephone call from the caller C's telephone **122**, out to the caller C's local loop **124**, across long-distance lines **126**, to the callee B's local loop **128**, and finally

to callee B's telephone **120**. This is an example of a telephonic device **122** to telephonic device **120** call.

If the non-subscriber callee C leaves a voice mail message for the subscriber B, the subscriber B may have the ability to "call" the system **100** from an outside line (*e.g.*, from telephone **120**) over the PSTN **102** and receive his voice mail, as is the conventional voice mail practice. Preferably, the subscriber B would dial his or her subscriber number and use a touch-tone keypad to navigate through various levels of electronic voice mail command menus to retrieve his messages.

Because a call may be placed from an IP device or from a conventional telephone, and a call may be directed toward an IP device or switched out to the PSTN, the present invention preferably allows four types of completed calls and data communications: (1) IP device to IP device; (2) IP device to telephonic device; (3) telephonic device to IP device; and (4) telephonic device to telephonic device.

In addition to voice traffic, the IP communications system **100** of the present invention also preferably supports facsimile and other data communications transfers. For example, the network node **114**, **116** may have the capability automatically to detect the "squelch" or calling tone (CNG) of an incoming fax transmission and return the proper tones to the sender to set up a fax transmission. The network node **114**, **116** preferably directs the incoming fax to a database for temporary storage

and notifies the subscriber that a fax is waiting on the system for retrieval.

2. Detailed System Architecture

FIG. 2 generally depicts a sample system architecture for the present invention including the IP communications system **200**. The system **200** generally works in conjunction with a circuit-switched communications network **202**, such as the Public Switched Telephone Network (PSTN). The PSTN **202** is connected to a series of electronic communications devices **210-220** through various transmission mediums, both wired and wireless, included in the PSTN **202**. For example, **FIG. 2** includes a telephone **210** and fax machine **212** for sending voice and graphics over the PSTN **202**. There may also be a satellite dish **214** or other large-scale wireless communications systems. Wireless applications such as mobile telephones **215**, voice and text pagers **216**, and personal digital assistants **217** (PDAs) also utilize the conventional PSTN **202**. These wireless devices often include a system of wireless "cells" **218** for routing and relaying calls to and from the PSTN **202**. Finally, **FIG. 2** shows a computer system **220** which may be connected to the PSTN **202** by way of a wired or wireless modem or other communications connection.

As described above, the PSTN **202** includes many "local loops" or local telephone systems that are connected together either directly (*e.g.*, with a router or other gateway) or through high-speed long distance lines (*e.g.*, fiberoptic cable or satellite links). By interconnecting

the local loops, the large scale PSTN **202** seamlessly routes telephone calls and other communications across large geographical distances. For example, a user in one city may place a call from a telephone connected to his or her local loop (owned and/or operated by Company A) to a second user whose telephone is directly connected to a second local loop (owned and/or operated by Company B) in another city. That call may be switched from Company A's loop, across a third company's long distance lines, and finally to Company B's local loop. Because of the interconnectivity, the call switching appears seamless to the caller and the callee.

The lower half of **FIG. 2** (the network nodes **240** and below) expands the communications capabilities of the PSTN **202** by interconnecting the PSTN **202** with a packet-switched network **222** such as the Internet or a local intranet. This additional packet-switched network **222** preferably works in communication with various servers (*e.g.*, **226**, **228**) and databases **224** to form an IP communications system **200**, capable of both transmitting information along the packet-switched network **222** and transferring information out to and in from the circuit-switched PSTN **202**. The IP communications system's servers and databases may be interconnected via a transmission medium such as a high bandwidth Ethernet pipe **244** which is also connected to the packet-switched network.

One of these servers, the gatekeeper **226**, is preferably connected to the public Internet or other packet-swtiched data

communications network **222**. The users of the IP communications system **200** preferably access the system by logging onto the Internet **222** through their Internet Service Provider (ISP) or other Internet connection mechanism (generally indicated as the IP cloud **222** in **FIG. 2**). The gatekeeper **226** has the functionality to identify these users on the Internet **222** and reroute their communication along the proper communications path.

The IP communications system **200** architecture also generally includes one or more network nodes **240** and/or SS7 gateways **242** connected between the IP network **222** and the PSTN **202**. Because these networks **202**, **222** are of two different types, (*i.e.*, circuit-switched and packet-switched) the network nodes **240** act as interpreters or interconnection computers that are capable of translating and rerouting signals to and from a circuit-switched network **202** (such as the PSTN) and a packet-switched network **222** (such as the Internet). The SS7 gateway **242** may be an interface between the network nodes **240** and the circuit-switched network **202** and may assist the network nodes **240** with call setup and signaling controls. Although much more detail will be given below, the network nodes **240** generally allow the interoperability of the two disparate network types **202**, **222**.

A subscriber to the IP communications system **200** is generally able to place and receive telephone calls from his IP device (*e.g.*, **238**) which includes system client software. The software client preferably runs on a multimedia IP device **238**, which may include a CPU,

and "multimedia components" such as a sound card, microphone, speakers, and/or a headset. These multimedia components aid the user in sending and receiving voice signals through the IP device **238**. The multimedia IP device may be a stand-alone personal computer (PC) **238**, a workstation, a wireless personal digital assistant (PDA) **239**, an electronic mail device or other web appliance **236**, or any other electronic communication device capable of performing telephony functionality according to a predefined call standard such as H.323, SIP, MGCP or other standard.

The IP device client **238** also includes client software, installed from a disk or downloaded from the Internet, that allows the subscriber to use the IP communications system **200**. The client software enables communications between the IP device **236**, **238**, **239** and the gatekeeper **226**, and provides the user with an intuitive graphical user interface (GUI) for utilizing the system. The client software may also provide certain functionality such as: call forwarding; call barring ("do not disturb"); direct callback; auto-registration; auto-upgrade; auto-verification of IP address; call "accept;" call "reject;" voice mail, fax and missed call notification and other features which are described in more detail below.

To log onto and utilize the IP communications system **200** the user preferably has a connection to the public Internet, a local intranet, or some other packet-switched communications network **222** that is also accessible to the gatekeeper server **226** of the IP

communications system **200** (*e.g.*, by way of a high bandwidth Ethernet pipe **244**). This dial-up or other Internet connection is generally depicted as an "IP cloud" **222** in **FIG. 2**.

The gatekeeper server or "gatekeeper" **226** acts as the connection interface between the subscriber of the system (accessing the system through his or her IP device **236**, **238**, **239**) and the remainder of the components (servers **228**, databases **224**, network nodes **240**) of the system **200**. The gatekeeper **226** is connected to the IP cloud **222** through a broadband communications medium, such as a fiberoptic cable **244**. This communications medium is preferably a high bandwidth Ethernet pipe **244** as indicated in **FIG. 2**.

The network nodes **240** also preferably provide the voice mail facility so that when the callee is not available, the caller can leave a message for him or her. Each user who subscribes to the voice mail service is allocated a personal voice mail box. Users may call the network nodes **240** to access their voice mail messages. Faxes received by the IP communications system **200** also may be stored and transferred via FTP (file transfer protocol) by the gatekeeper **226** to the software client upon request (user gets notification that a fax is waiting).

The network node(s) **240** are the communication interpreters that allow the packet-switched network **222** just described to communicate seamlessly with a general circuit-switched communications network such as the PSTN **202**. The network nodes **240** are generally

capable of transferring both real-time voice communications and non-real-time messages (such as voicemail and/or facsimile) from one form of communications network (*e.g.*, circuit-switched) to another (*e.g.*, IP) and vice versa. The network node **240** may be redundant and scalable in that more than one network node **240** may be connected to each local loop or PSTN trunk.

The network node **240** hardware may include a high-end CPU such as is common in personal computers and/or workstations, hard disks, memory (RAM, ROM), as well as several IP telephony cards. Each telephony board preferably interfaces with T1/E1 PSTN trunk lines and a 100 base T interface (on the IP network **244** side). In a preferred embodiment, the network node includes the NMS CG6000 telephony board which allows for, at a minimum, 120 calls to be processed by the board at once. Each of the network nodes **240** may include four telephony boards for a total of 480 ports or 480 simultaneous inbound and outbound VoIP calls. If more ports are needed, additional network nodes **240** may be deployed. Preferably, the network node **240** will autodetect additional CG6000 boards (or other telephony boards) if they are present in the system. By default, the system **200** may automatically utilize all existing resources on every board. Alternatively, certain boards or ports may be reserved for emergency use or some other purpose.

With multiple boards, each network node **240** preferably supports up to 480 simultaneous calls. Voice over IP (VoIP) functionality is built into every port with a standard IP call control protocol such as

H.323, MGCP or SIP. Supported codecs, for example, may include G711, G723, G726, and GSM.

The network nodes **240** also contain software that gives functionality to the VoIP telephony boards. The network node software, developed as part of the present invention, generally includes the following features: multi-threading core engine; telephonic device-to-IP device and IP device-to-telephonic device connectivity; interface to software client, subscriber database, gatekeeper databases, and notification server; voice/fax mail; IVR; network node-PSTN signaling; security control; and watchdog function. The network nodes **240** are preferably designed for scalability, portability, and ease of maintenance, so that the nodes can be serviced while still operational in the field. In one embodiment, the nodes **240** are developed using a common workstation operating system. The nodes **240** preferably support customizable call flow through a built-in scripting engine. The code base of the network nodes **240** preferably contains no operating system-dependent code, enabling the nodes **240** to be easily ported to another OS, as long as the NMS, API and RadVision H.323 API is supported by the new operating system.

Multiple other servers and computers may also be connected to the IP communications medium **244**. These other servers and devices, as well as those already mentioned, may be implemented on the same computer, various separate computers or any subcombination thereof. For example, a redundant database engine **224** preferably exists as part of a

computer connected to this medium **244**. The redundant database engine **224** includes several databases that keep track of calls, subscriber identifiers, subscriber passwords and user preferences. The database **224** also preferably keeps track of the current (dynamic) IP address for system subscribers. Therefore, as subscribers log on and off of their ISP, changing their dynamic IP address, the system **200** updates its database records. This "dynamic" IP address for a current user is matched to the subscriber identifier for that user in the database. This dynamic IP address to subscriber identifier association in the system database preferably allows for subscriber authentication and call re-routing from the IP-switched network to the PSTN (explained in more detail below).

There may also be a media server **228** capable of storing and/or streaming multimedia information to the user in various circumstances. For example, the media server **228** may be able to store and play voice data in an electronic format such as ".wav" files. At a later time, the ".wav" files may be sent or streamed to the software client to allow the user to hear the voice mail or other multimedia messages. The media server **228** may also support other multimedia formats for audio (*e.g.*, .mp3, .wav, .amf, and .rm) and video (*e.g.*, MPEG, QuickTime). The media server **228** may be implemented on a conventional PC with software.

The system **200** may also include one or more "administrative" elements that aid the system administrator in tracking and monitoring system performance. These elements may include an On-

line Monitoring Computer (OMC) **230** and/or a network manager **232**. The OMC gateway **230** is preferably a piece of software that monitors the performance of all of the individual network nodes **240**. The OMC gateway **230** typically tracks the network nodes **240** and tracks the number of calls each subscriber places and receives, the duration of these calls, how many trunks are available for additional calls to be placed, and other administrative features of the network. This traffic information may be useful in determining what percentage of system resources are generally being used, and may help the system administrators to determine the appropriate times to upgrade or replace hardware and software.

The network manager computer **232** may be the configuration element of the system **200**. The network manager **232** may monitor the network node **240** or other system components, or receive monitored information from the OMC gateway **230**, to determine at what point the system **200** is running at or near full capacity. When near full capacity, the network manager **232** may signal to the network nodes **240**, **242** to cease accepting new calls. This network manager **232** may perform any number of administrative, configuration, and monitoring tasks.

The message notification server **234** preferably logs and tracks incoming calls and messages when the user is not able to be located. When the user logs onto the IP communications system **200**, the message notification server **234** typically sends a message to the client software (on **238**) that indicates how many and what types of messages

are awaiting the user. For example, the notification server **234** may indicate that a caller called and did not leave a message -- a missed call event -- (indicating the caller's number or subscriber identifier). The notification server **234** may also indicate that a fax or voice mail was received and is waiting to be "picked up" by the user. Finally, the notification server **234** may indicate that a call was received and the caller left a voice mail message. These indications are useful for the user because they give the user a complete indication of activity since the last time the user logged onto the IP communications system **200**. The client software may provide further choices of future user actions based on these received notifications (*e.g.*, automatic call back, fax/message retrieval).

The more complex elements of the IP communications system **200**, the gatekeeper **226**, network node **240** and software client (on **238**), will now be discussed in more detail. **FIG. 3** divides the network node architecture into functional elements. At the bottom of **FIG. 3**, the "network interface" **310** represents the physical layer or interconnection between the packet-switched network and the PSTN. Commonly, this physical network interface **310** will be the telephone line cards of the network nodes (*e.g.*, CG6000 telephony boards).

Above the physical network interface **310** is the core software engine **312** for the network interface **310**. This core software engine **312** generally includes the various software drivers and other functionality to control the telephone line cards and telephony boards. For example, the core software engine **312** may provide individual controls for VoIP

compression/decompression, for data encryption/decryption, to establish and send voice prompts, and to control other features of the telephony boards.

Above the core software engine **312** is another software layer **314** that preferably includes third party software packages and routines that enable IP communications functionality. For example, there may be the functionality to utilize various network protocols such as R1, R2, ISDN, and/or SS7 **316**. There may also be functionality to enable voice/fax switching **318**. There is preferably a VoIP compression device with real-time fax capabilities **320**. The system may also include a series of recorded voice prompts **322**, such as “voice mail not available,” that the system may play for a caller or a user under certain circumstances.

There may also be an IP sockets/RTP/RTCP module **324** that allows quality voice signals to be sent over the packet-switched network even though the voice signals may be “bursty” by nature. This module **324** may include a buffer and sorter that reorders the sent packets as they are received and sends these packets as a datastream to the user. Finally, there is preferably a module **326** that allows access to the various system databases. For example, as previously described, there is preferably a subscriber database that allows the tracking and logging of subscriber activity and preferences.

The top layer **330** of the network node functional diagram (**FIG. 3**) is the software-based virtual intelligent agent application. The

virtual intelligent agent software **330** integrates all of the other software (*e.g.,* **312**, **314**) and hardware (*e.g.,* **310**) found in the network node together to enable system functionality.

FIG. 4 displays an additional functional diagram of the IP communications network architecture as found in at least one embodiment of the present invention. This diagram generally indicates two network nodes **400** connected to one "backend server" **405** which includes one or more gatekeepers, media servers, databases, and other IP system components. As described above, there may be any number of network nodes **400** connected to the PSTN or to a particular local loop in the PSTN, and there may be more than one backend server **405** at each location. Preferably, there is at least one backend server **405** and at least one network node **400** connected to each local telephone loop to which subscribers to the present invention are connected. As more users utilize a certain part of the system, additional network nodes **400** may be inserted (usually on the fly) to service these additional users. The additional network nodes **400** would preferably include additional trunk cards **410** for communicating with the PSTN or other circuit-switched network.

The backend server **405** includes most of the IP-side functionality of the system. The backend "server" **405** may be just one computer with assorted functional features, but the backend server **405** is preferably a combination of servers, databases and other devices that are connected by a high speed communication medium such as a high

bandwidth Ethernet pipe **412**. In **FIG. 4**, the backend server **405** includes the gatekeeper (GWSVR **415**), a mailbox manager **420** (notification server) and a REAL media server **425** (or other multimedia server) which enables the streaming of multimedia information to the software client.

The backend server **405** also preferably includes one or more databases **430** (DB) that include information for tracking the accounts, preferences, dynamic IP addresses and calls of the subscribers. There may also be a connection database **435** that aids in other administrative tasks. The backend server **405** preferably includes SMTP **440** or other electronic mailing functionality to aid the user in the storage and transmission of electronic messages via email. There may also be an FTP module **445** or other file transferring module that enables the transmitting of a received facsimile or other electronic document to the user on his or her software client.

This backend server **405** is preferably connected to one or more network nodes **400** through a high speed communications medium such as a high bandwidth Ethernet pipe **410** (which is typically connected to the Internet). The network node **400** consists of both hardware and software that allows the packet-switched, IP-switched network to communicate with a circuit-switched network such as the PSTN. The network node **400** includes one or more trunk cards **410** that provide physical communication between the networks.

The network node **400** also preferably includes a database service **455**, a switching service module (SWI) **460** and a PSTN call setup service (ADI) **465**. The database service **455** allows the network node **400** to interact with the various databases (*e.g.*, **430**) in the backend server **405**. The SWI **460** and ADI **465** services are common software interface modules.

As described above, the gatekeeper facilitates the interconnection and communication between the software client connected to the public Internet, local intranet or other IP-based network, and the network node connected to the PSTN. This functionality provides not only a communication and interpretation methodology, but also facilitates many aspects of the present invention.

The gatekeeper is the initial server entity with which the client's own IP devices and network nodes interact. Client logons/logoffs are performed via the gatekeeper. Clients that are logging on have to supply their subscriber identifier and password. During the login process, the gatekeeper authenticates the user and registers the client dynamic IP address in the database. In this way, the system has a database record that matches a user's subscriber number to the user's dynamic IP address during that particular IP session. Typical users of the system will access the Internet through an ISP and be assigned a different IP address during each session. The gatekeeper tracks these dynamic IP addresses. Upon logging off, the gatekeeper "unregisters" the client's IP address from the database. Basically, the gatekeeper keeps track of "where" (in the IP

address world) you are so that the system can ring you appropriately when calls are received. If a call is received and no dynamic IP address is found for the callee subscriber in the database, the system will know that the callee is not logged into the IP network and may try to transfer the call out to the PSTN or prompt the caller to leave a message, according to the callee's predefined preferences.

The gatekeeper also allows the clients to query for missed calls, voice mails or faxes; it supplies the necessary information in the event of a retrieval. Clients further initiate outgoing calls through the gatekeeper. The server will query the database on behalf of the calling client and will supply it with the IP addresses of the destination client and gatekeepers. Apart from these main communications functions, the gatekeeper may also assist the users in changing certain user options (preferences), in upgrading the client software, and other administrative tasks.

The gatekeeper handles requests from clients using a fully multithreaded model. It is preferably capable of serving simultaneous requests from multiple clients at the same time. Multiple gatekeepers may be deployed in clusters to handle heavy loads as well as to improve server redundancy. Therefore, if one gatekeeper malfunctions, a redundant gatekeeper server can pick up the extra user load. Also, these gatekeepers need not physically exist near one another. The servers are preferably all in communicative contact with each other through a high speed transmission medium, and the gatekeepers may exist in various

places around the world to better serve local customers of the IP communications system.

A more detailed description of the network node architecture is in **FIG. 5.** At the top of the network node architecture diagram (**FIG. 5**) is a user interface module **505**. The user interface **505** preferably provides an outside environment for developers to upgrade or add functional additions to the IP communications system. This user interface **505** may be both a physical point of access into the system for add-ons, or a software socket for new program information. Preferably, these add-ons and developments may be incorporated into the network node while the network node is operational.

There may also be a virtual intelligent agent module **510** that provides functionality to the various telephony boards and software modules.

The network node application section **502** of the network node architecture diagram (**FIG. 5**) may also include several functional software and/or hardware modules. For example, there may be an IP call management module process **515** that rings a user's IP device upon receiving an incoming call. There may also be an RTP/RTCP stream management module **520** that allows an uninterrupted voice stream to reach the user even though the Internet is packetized and bursty in nature. This occurs even under heavy Internet traffic conditions. This

module **520** preferably handles the QOS streaming which can ensure a quality voice signal in such a bursty environment.

The application section **502** of the network node may also include an address translation module **525** that handles the registration of an IP address to its corresponding user. This module **525** may authenticate or validate a user by checking his or her current IP address against a stored IP address for that user. Generally speaking, when an incoming call reaches the network node, it will locate the gatekeeper which will look up the dynamic IP address from the database for that user, check to see if the user is logged on, and then authenticate the user against the stored IP address. Finally, the application side **502** of the network node may also include an IVR function management and PSTN **530** call management module that retrieves the stored voice mails and faxes for a user when they log on and directs the flow of all the stored prompts and messages. More will be said about this below.

The network node also has software developed using the Fusion SDK (Software Development Kit) or other software development tools **504** that includes modules that facilitate interaction with the Internet/intranet (as packet data **583**) as well as the PSTN (PCM data **588**). The Fusion SDK-developed modules **504** may include an IP call control module **540** including an H.323 stack **535**. This module **540** enables the network node to setup and place a call as well as to receive incoming calls. The Fusion SDK layer **504** may also include media stream protocol processing TX board APIs and host stream APIs **545** to facilitate

the transfer of various data packets **583** over the IP network **585**. These modules **545** preferably handle all of the subscriber ports associated with the network node's interconnection with the IP network **585**. This module **545** may include data transport (RTP/RTCP) **550** as well as resource configuration (subscriber ports, host endpoints) **555**. This port is communicatively connected to the Internet/intranet or other packet-switched network **585** to which users (clients) are connected.

The final section of the Fusion SDK-based applications **504** includes the CT access and CTA TRAU service APIs **560**. These modules **560** generally allow for call control and ending over the PSTN network. This module **560** may include an H.100/MVIP switching module **565** which is the engine that allows the "hot plugging" of components while the system is running. This module **560** may also include data conversion vocoders/RT fax modules (codecs) **570**. This module **565** may also include PSTN call control **575** and IVR control functionality **580**. Generally, these modules as a group allow the network node to communicate with and send PCM data **588** over a circuit-switched network such as the PSTN **590**.

As described above, these network nodes are preferably distributed over various geographic regions to better serve various subscribers. Preferably, there is at least one network node connected to the local telephone switch (local loop) to which each subscriber to the system is connected for regular telephone service. With this dispersed network node architecture, subscribers in various geographic regions will

be able to utilize the system fully (*e.g.*, with an IP device to IP device call) without necessarily interacting with the circuit-switched network.

FIG. 6 details a functional diagram of the hardware and software on the multimedia software client. At the highest level, the software client includes a user-friendly graphical user interface **600** that aids in the placing and logging of telephone calls and faxes. This interface software preferably runs in a Win32 API environment which may include MFC and customized classes **605** developed as part of the present invention.

This client architecture also preferably includes functional modules that allow communication between the client IP device and the gatekeeper over the Internet, intranet, or other IP-switched network. For example, there is preferably an H.323 call manager **610** that handles all of the inbound and outbound call routines and the RTP/RTCP data transports. There may also be a subscriber manager module **615** that interacts with the gatekeeper to determine whether or not a user with a particular IP address is logged on. There may also be one or more utility modules **620** that provide ease-of-use and functional aspects to the client software.

The H.323 call manager **610** is preferably connected to an H.323 call stack API **625** and wavein/waveout threads **640** that control the flow of packet data **690** over the IP network **695**. For example, there may be an IP call control module **630** and an RTP/RTCP data transport

module **635** that facilitates this data communication. Preferably, there is a data buffer between the H.323 call manager and the H.323 call stack to make sure that the "bursty" voice packets have time to be properly re-aligned so that the sound is continuous. Because IP packets may arrive out of order and at different times, the data packets are preferably delayed on the order of hundreds of milliseconds to give the system time to rebuild the voice stream.

The wavein/waveout threads **640** may include a G723 codec module (coder/decoder) **645** and a wave device manager **650** that allows the voice signals in wave format to be sent and received. The user interface may also include a signal display so that the user can view the shape of the voice signal.

The subscriber manager **615** is preferably connected to a socket thread **655** that directs the communications with the gatekeeper **660**. These communications preferably take place using the TCP/IP protocol **692**. The communications include the gatekeeper's query as to the client's IP address as well as whether or not the client is currently logged into the system.

The utility module **620** is preferably connected to a variety of customized control modules **665** including an inbox **670**, a phone book **675**, a configuration module **680**, and an auto-upgrade module **685** that checks to see if a newer version of the client software is available on the

system. These functionalities will be discussed in more detail with respect to the user interface.

FIG. 7 details one example embodiment of a graphical user interface (GUI) **700** for use with the present invention. The GUI **700** generally provides a subscriber to the system with an easy and efficient interface through which to take advantage of the various features of the present electronic communications systems. The subscriber generally navigates around and provides information to the GUI **700** by way of a computer mouse, keyboard or other input device. The GUI **700** is preferably part of the client software running on the subscriber's IP device.

The GUI **700** may generally include a system identifier (*e.g.*, a telephone number) window **705** which displays the subscriber identifier of the other party during incoming and outgoing calls. There may also be a drop down menu which displays a call history, or recent called/calling subscriber identifiers **710**. This list may include the functionality to allow the user to select the number with the mouse pointer and automatically call these other parties.

When placing an outgoing call, the subscriber preferably types the subscriber identifier or telephone number in the display window **705** with the keyboard or by selecting the numbers on a virtual keypad **725**. The subscriber may preferably choose to dial **715** the number or clear **720** the window **705** to type in a different subscriber identifier.

The GUI **700** also preferably contains certain status information about the subscriber's current session. For example, there may be a status bar **730** indicating the login status and subscriber identifier of the current system session. There may also be a bank of indicators that notify the user how many missed call events **735**, voice mail messages **745** and/or facsimiles **745** are waiting for the subscriber in his or her inbox. These indicators are preferably updated by the notification server when the user logs into the system. The indicators **735**, **740**, **745** generally provide the subscriber with status updates.

The GUI **700** may also include one or more standard "WINDOWS" functions that allow the user to manipulate the GUI **700** in the same way as other graphical software. For example, there may be an "exit" button **750** or a "minimize" button **752**. There may also be a "help" button **754** that provides the user with immediate or on-line information to aid the user in using the GUI **700** and the communications system in general.

The GUI **700** may also have an address book function **764** that allows the user to store frequently called numbers and subscriber identifiers. The entries in the address book **764** preferably allow the user to store the name, subscriber identifier or telephone number, and other information about each entry in the address book. Using the address book **764**, the user may be able automatically to call a subscriber or other party by "clicking" on the party's name or identifier in the address book **764** list.

The system may also have an inbox **762** which stores all of the voice mail messages, facsimile messages, and/or missed called events in one combined list of information. Preferably, the inbox **762** details what type of event was received, when it was received, from whom it was received, and the length of the message, if any. To retrieve a received voice mail message or facsimile, the subscriber preferably need only "double-click" or otherwise select the entry in the inbox **762**. The client software then preferably opens up the appropriate fax viewer (.tif viewer) or streaming media player to view the fax or listen to the voice mail message. These messages may be replayed, stored, forwarded and otherwise manipulated by the subscriber.

The inbox **762** may also have an automatic callback functionality for the entries in the inbox **762**. For example, to call back a person who called but did not leave a message (a missed call event), the subscriber preferably need only double-click or otherwise select the missed call event from the inbox **762** list of entries. This same callback functionality may also enable the automatic callback of parties who leave fax and voice mail messages.

The GUI **700** may also provide the functionality for users to define certain system preferences from the GUI **700**. For example, there may be a series of mouse menus (or other selectable features) that may be chosen and altered by the subscriber. For example, "right–clicking" on the GUI **700** or otherwise selecting to see a menu of options, may provide a selection menu **770**. This menu **770** may allow the user to logoff, exit, or

select from a further list of utility button options and/or online setup options.

The utility button options **775** may provide the subscriber with certain use and upgrade options. For example, the subscriber may be able to access and edit his or her phonebook and inbox from the menu **775**. The subscriber may also be able to automatically look for a newer version (upgrade) of the client software on the Internet, or the subscriber may be able to find further information about the current version of the client software. Finally, there may be one or more "user configure" options that allow the user to change login options, the facsimile data storage path, or other configurable options.

The online setup menu **780** preferably includes various options to configure and set up the GUI further in accordance with the subscriber's preferences. For example, there may be a volume adjustment or the ability to change the subscriber password (or PIN). There may also be a "do not disturb" button that disables the "ringing" feature of the client software. With "do not disturb" enabled, the subscriber may still see the notification indication about incoming call events, and the subscriber may also be able to place outgoing calls and facsimiles, but the subscriber will preferably not be bothered by incoming calls. The incoming calls will preferably be routed directly to the subscriber's voice mail or other messaging function.

Finally, the online setup menu **780** may allow for "auto silence detection" during voice calls. Because there is always some background noise during a telephone call, even when no party is speaking, the system may normally send these "hisses" across the communications system. Because the parties normally will not intend for this background noise to be sent, it is a waste of system bandwidth to send these packets. With auto silence detection, the system may stop sending packets if a predefined amount of time elapses during which the system detects only this background noise. Once a subscriber does begin to speak again, the system will preferably start sending data packets again.

The general components of the present invention are preferably designed to be scalable, redundant and adapted for rapid deployment and upgrading while the system is running. Components are preferably designed using industry standard technologies such as Signaling System 7; TCP/IP for data communications; and H.323, SIP or MGCP for call setup and control. The scalability will provide for seamless addition of new services. As current network nodes and server components run near their maximum throughput, additional nodes and servers may be added to the system. As the new systems are added, all of the nodes and servers should be programmed to share the load of data transfer more evenly, rather than fully using one server while another stays idle nearby. These components should also be rapidly deployable in that additional components can easily come on-line while the system

continues to operate. This may be facilitated by using a programmable state machine using a simple scripting language (*e.g.*, LUA).

The components also preferably utilize open architecture components wherever possible. By opening the architecture, the potential for additional, "value-added" services to be generated is greatly enhanced. With an increased number of developers involved, the system will gain popularity and use more quickly. Sources of open architecture features including using an industry standard bus (*e.g.*, compact PCI), using the Linux operating system or other open source software for backends, running the gatekeepers on Solaris workstations, and utilizing ACE libraries.

Although not essential, the servers and other equipment may also be placed in front-open (easy access) cabinets with both ESD and EMI protection circuits used. Preferably, even the standard computer components, such as disk drives, power supplies, and trunk cards, are all "hot pluggable" in that they can be removed, upgraded, and/or replaced without powering down the system or the system component.

The system may also include custom call scripting that facilitates user-configuring of various features of the IP communications system. For example, a user may script an icon to play a message (*e.g.*, "I'll call you right back."). This scripting may also allow the user to change his/her voice mail announcement.

3. Methodology Examples

The above hardware and software descriptions schematically divide the present invention into functional components. To aid in comprehension of the present invention, several exemplary "scenarios" of use of one embodiment of the present invention will now be detailed. As above, these scenarios are for purposes of example only, and should not be used to limit the present invention.

The first example describes a person registering to use the IP communications system for the first time and thereafter receiving a call from another person. The user preferably has an IP device running a common operating system such as Windows 98, NT, 2000, ME or CE. The user may register for the system, either by telephone or through the World Wide Web (the "web"). The IP communications system will assign the user a unique subscriber identifier, which is often a telephone number (VIA number or "subscriber number") and a PIN (personal identification number) to be used as a password. Because the present invention can be used with both the PSTN and the IP-switched network, the subscriber number assigned to the user preferably corresponds to the format of telephone numbers in the PSTN where the user resides (*e.g.*, numbers in the U.S. would have a corresponding area code). This facilitates seamless switching between the IP-switched and PSTN networks. The PIN may preferably have at least six digits and is used for authentication and security.

After being assigned a subscriber identifier and a PIN, the user preferably installs the client software from a disk or installs the software directly from a web page sponsored by the IP communications system. Alternatively, the IP device may come with the client software preinstalled. The installation program may use an auto-install aid, such as an "Install-Shield Wizard" for use with common operating systems. When the user wishes to log onto the system, the user will start the program on his or her IP device at which time the user interface will appear on the user's display.

During this initial logon, the system will prompt the user to input his or her subscriber identifier and PIN. The client software includes a registration table that directs the software client to a specific gatekeeper from which the user can log onto the IP communications system. This gatekeeper is preferably connected to the Internet, a local intranet, or some other IP network to which the user is also connected. Most often, the user will use an Internet Service Provider (ISP) to connect to the public Internet with which the gatekeeper will also be connected. Preferably, the user will always be connected to the system through the same gateway in future sessions.

Based on the registration table, the client software will send a message to the appropriate gatekeeper including the user's subscriber identifier and PIN and request that the user be logged into the system. The gatekeeper queries the database to determine if the proposed subscriber identifier and PIN represent a valid subscriber. If the

gatekeeper named in the registration table is not operating properly, there may be additional gatekeepers daisy-chained together the handle the database query. This is one of the many redundancies that may exist in the present system. Once the gatekeeper verifies the validity of the user, the user is logged onto the system. The gatekeeper will also query the software client for the dynamic IP address of the IP device on the packet-switched network and enter the user's current dynamic IP address (which typically changes each time the user logs into the packet-switched network) into the database for use by various features of the system.

The network nodes, that sit between the PSTN and the Internet, contain telephony boards with ports to handle telephone and IP calls. The local PSTN exchange is programmed such that it will route calls placed to the user's subscriber identifier to the appropriate network node, or a series of network nodes coupled together. This automatic routing is the reason the subscriber identifier should correspond to the particular dialing practices of the public telephone system in the locality of the user.

When a caller places a call from a conventional telephonic device to the user's subscriber identifier (number), the call gets routed by the PSTN to the appropriate local loop of the PSTN where the user resides. At this point, the network node attached to the local loop takes the call off of the PSTN network and requests the gatekeeper (assigned to that subscriber identifier) to check the database to determine if this user is currently logged into the packet-switched network. If the gatekeeper

finds that the user is logged onto the system, the gatekeeper will look up the user's current dynamic IP address and perform an authentication socket call to the user's software client. In this socket call process, the gatekeeper will request the subscriber identifier and PIN (password) from the software client. If the software is running on the software client, the software will activate itself upon receiving the socket call and determine if the proper subscriber is on that particular client (*i.e.*, authenticate the subscriber identifier and PIN). The client software will then send a confirmation or acknowledgment back to the gatekeeper of the user's identity and validity.

This confirmation performs an authentication process to prevent unauthorized users from using the system on a software client. For example, if a subscriber is using the system and forgets to logout of the system after a session, and another person thereafter logs onto the Internet and runs the client software on the subscriber's IP device, the client software will not allow the IP device to ring when called because the "new" dynamic IP address (assigned to the second user) will not match the IP address stored in the system's database (from the subscriber's last session). The client software will notify the gatekeeper that an incorrect user is logged into the system.

After the registration/authentication process, the gatekeeper signals to the network node that the user is logged in, and the network node sets up an H.323 (or SIP, MGCP) call session for the user. The network node will setup an H.323 call with the client. This H.323 session

notifies the client of the incoming call and identifies the subscriber number or other identifying information about the caller. The client receives this information, pops the software application onto the user's display, and signals the user through his IP device (*e.g.*, with an audible "ringing" sound or vibration). The software application window will preferably identify the caller and inquire as to whether the user would like to accept or "reject" the incoming call (*e.g.*, using mouse-based control buttons). This effectively performs a call-screening function.

If the user elects to accept the call, for example by clicking with a mouse on a dialog button, the client software will signal the network node that the call has been accepted, and the network node will set up a call session between the caller's telephone and the user's IP device. The network node effectively acts as an interpreter or data converter during the call session. The voice data comes into the local telephone system as voice data on the PSTN, and the network node converts the voice data into packet data to be sent to the software client (over an IP-switched network). The client software will collect the data packets, sort the packets into their correct order, and play the caller's voice on the software client (utilizing the sound card and speakers of the software client). The network node preferably performs data compression and decompression (coding/decoding) while packetizing the voice information for distribution to the software client. This compression/decompression allows compressed data to be sent over the

network thereby allowing a low bandwidth connection to provide high fidelity recreation of the caller's voice signal.

The client software has additional buffering capabilities utilized when receiving the data packets from the network node. As the packets of voice information are received by the client software, they may arrive at the client in a different order than they were sent by the network node. The software client preferably utilizes a "jitter buffer" or other buffering technology to receive the packets (in any order), and analyze the packet packaging bytes to determine the proper order of the packets. Each packet can therefore be placed in the proper order, and, after a slight time delay, the stream of reordered packets can be played for the user without interruption. This buffering may impart a delay of up to approximately several hundred milliseconds.

If, during the authentication process, the gatekeeper is unsuccessful in finding the user logged into the network (because the user is either not logged into the Internet or the client is unavailable for some other reason), the gatekeeper and the network node will enter "call completion mode" and determine how to handle the call. In one embodiment of the present invention, the gatekeeper will signal to the network node that the caller is not available, and the network node will play an audio message through the PSTN to the caller indicating that the callee is not available. The message may request that the caller leave a voice mail message for the callee or the message may provide further information (based on the user's preferences and pre-recorded voice

prompts which are preferably stored in the network node or an accompanying database). If the caller chooses to leave a message, the network node will record the voice message and preferably store the voice message in the database for future playback by the callee. The network node will also preferably signal the notification server to notify the user that a voice mail message is waiting for the user in the database. In this way, the next time the user logs onto the system (and the notification server is activated by the gatekeeper), the user will be alerted that a voice mail message is waiting.

The notification server may be programmed with several different notification options. Preferably, the user is able to set these various options using the client software. For example, the notification server may wait for the user to log back into the system before notifying the user via his IP device. The notification server may also be programmed to call the user on his mobile telephone, text or voice pager, PDA, or some other electronic communications device.

The notification server may also be set up to forward the call rather than prompt the caller to leave a voice mail message. For example, when the gatekeeper determines that the user is not logged into the system, the notification server may signal to the network node that the call is to be forwarded to the user's mobile or land-based telephone number. Therefore, without the caller performing any additional steps, the call may be forwarded to this outside phone number. The network node will preferably forward the call back out to the PSTN and ring the

user at his desired forwarding phone number. In this way, although the caller called the user on a number identified with the IP communications system, the user may receive the call on his mobile or other phone or pager, all preferably without the caller being aware of the forwarding process. This seamless integration of the circuit-switched PSTN and the packet-switched IP network may provide a more convenient way to reach callees.

If the user cannot be found and all call forwarding options have been exhausted, the caller is preferably prompted to leave a voice mail message. If the caller chooses not to leave a voice mail message, or if the call is dropped before completion, the notification server is preferably programmed to notify the user that a "missed call event" has occurred. Although no voice mail message is waiting, the notification server preferably indicates to the user, as part of the client software, that a call was received but not completed since the user was last logged into the system. The notification server preferably includes the telephone number or subscriber identifier of the caller to aid in the user's ability to call the caller back at some time in the future. The callee's client software may also have an automatic callback feature, allowing the callee to initiate an immediate callback.

Similarly, if the caller sends a fax to a subscriber of the IP communications system, the caller will send the fax according to standard faxing procedures. As the fax call is routed to the subscriber's number, the PSTN again recognizes the subscriber number as being mapped to a

particular network node (or series of network nodes). When the network node receives the fax call, the network node "listens" for several seconds when the call first comes into the network node. If the network node receives a fax tone, *i.e.*, the series of squelches or CNG tones characteristic of an incoming fax, the network node will recognize the call as containing a fax message and will return the appropriate "return fax tone" to the caller's fax machine (as is common practice with conventional fax procedures).

After responding to the caller's fax machine, the network node will enter a "deposit mode" whereby the fax message is received by the network node and stored in the database for future downloading by the software client. After the fax is received, the network node preferably completes the call according to standard fax practices and signals the notification server to notify the user that a fax has been received. Accordingly, the notification server may determine if the callee subscriber is on the Internet or not, and is able to send a notification to the user via the client software. If the subscriber is on the system, or during the subscriber's next session on the system, the client software preferably has the ability to indicate that a fax was received by the system. The subscriber can then retrieve the fax via a .tif file viewer or other conventional fax viewing software.

In one preferred embodiment of the present invention, the method for sending an outgoing call from the IP communications system to another person is generally the reverse of the procedure just described

for incoming calls. The user types in, or otherwise enters, a destination identifier (either conventional PSTN number or subscriber identifier) to be called. The client software sends this call information to the subscriber's gatekeeper, and the gatekeeper may re-authenticate the subscriber to make sure that he or she is a valid subscriber.

The gatekeeper will then check the database to determine if the number being called belongs to a valid subscriber and whether, if that number belongs to a valid subscriber, that subscriber is also logged into the Internet by searching the database for a dynamic IP address associated with the subscriber number. If the callee subscriber is logged into the Internet, the gatekeeper will retrieve the callee subscriber's IP address from the database (which was entered in the database when the callee subscriber logged into the system). The gatekeeper will return the callee subscriber's IP address to the caller subscriber's client software so the call can be placed. Because both subscribers (caller and callee) are currently logged into the system, the call may be placed over the Internet (IP-switched), rather than having to go at least partially across the PSTN (circuit-switched). Therefore, the caller client software will preferably not go through the network node to place the call, because no translation from the IP network to the PSTN need take place. Instead, the caller's client software will setup an H.323 (or SIP, MGCP) call session to ring the callee subscriber at his IP address.

When the caller subscriber's client software sets up the H.323 session and rings the callee subscriber, the gatekeeper will again

attempt to authenticate the callee subscriber to make sure that the person at the designated IP address is in fact the person whom the caller subscriber is attempting to call. This authentication occurs by the gatekeeper performing a socket call to the callee client software and checking the response with the database information for that client. If the client software does not respond correctly (*e.g.*, the database had an incorrect or old dynamic IP address for the callee subscriber, the call is routed to the network node assigned to the callee subscriber. This network node enters "call completion mode" for the callee subscriber and determines how to handle the completion of the call. For example, as described above, the network node may transfer the call to one of the callee's PSTN numbers and ring the callee's mobile phone, landline phone or pager. Alternatively, the network node may play a "please leave a voice mail" message back to the caller subscriber. The callee's network node may then record the voice mail message to the database for future access by the callee, and the callee's notification server is preferably signaled to notify the callee subscriber when next the subscriber logs into the system.

A similar circumstance arises when a subscriber of the IP communications system calls another subscriber of the IP communications system, but the two subscribers exist on two separate, but interconnected, networks (A and B). In one preferred embodiment, the present invention facilitates a seamless interconnection of these two networks. In this case, the caller client software and the caller gatekeeper are preferably able to

locate an IP address for the callee, even though the callee is on a different network. For example, when the caller (on network A) attempts to make a call to a callee subscriber on network B, the caller client software will tell the caller's gatekeeper to locate the appropriate IP address for the callee. The gatekeeper will check the database and determine that the callee is a valid subscriber, but that the caller belongs to a separate network, and cannot therefore determine the callee's IP address. The gatekeeper will, however, determine that the callee's subscriber number belongs to network B, and the gatekeeper will be able to look up a DNS (Domain Name Service) address for network B. The gatekeeper preferably sends the network B DNS back to the client software which then accesses the network B DNS and asks for an IP address for the callee's number.

The client software will then attempt to contact the callee subscriber at that IP address and determine whether or not the callee is on the Internet. If the callee subscriber is on the Internet, the gatekeeper on the callee's network (B) can set up a direct H.323 session (or other call session) between the caller and the callee. If the gatekeeper on network B determines that the callee is not on the Internet or is somehow otherwise unavailable, the network B gatekeeper node will transfer the call to the network B network node in call completion mode. The network B network node will then determine how to complete the call (*e.g.*, call forwarding, missed call, voice mail) according to the practices outlined above. Any voice mail messages and other information is preferably stored at a

database connected to network B, where it can be retrieved by the callee the next time the callee logs into or calls the system. A notification server on network B will notify the callee.

A preferred embodiment of the present invention also allows a subscriber to the system to call a non-subscriber from their IP device using the client software. The call begins like every other call: the user dials the callee's phone number in the client software and the dialing information is sent to the caller subscriber's gatekeeper to determine if the callee is a valid subscriber. In this case, the gatekeeper will determine, through polling the database, that the callee is not a valid subscriber. The gatekeeper will then transfer the call to the network node and tell the network node to place a "regular" telephone call over the PSTN. Therefore, the network node will send the call out over the PSTN and ring the callee on his telephone (*e.g.*, mobile or landline) and complete the call as a regular telephone call. Throughout the call process, the network node acts as an interpreter between the PSTN and the packet-switched local network. To the subscriber, the call appears seamless, regardless of the whether the callee is a subscriber who is logged into the system, a subscriber who is not logged in, or a non-subscriber who is on a regular telephone.

The examples and embodiments of this invention can also utilize various wireless technologies, both IP-based and PSTN-based. Users from various disparate geographies can communicate in a limitless

array of circuit-switched and packet-switched schemes, all with near seamless integration.

Nothing in the above description is meant to limit the present invention to any specific materials, geometry, or orientation of parts. Many part/orientation substitutions are contemplated within the scope of the present invention. The embodiments described herein were presented by way of example only and should not be used to limit the scope of the invention.

Although the invention has been described in terms of particular embodiments in an application, one of ordinary skill in the art, in light of the teachings herein, can generate additional embodiments and modifications without departing from the spirit of, or exceeding the scope of, the claimed invention. Accordingly, it is understood that the drawings and the descriptions herein are proffered by way of example only to facilitate comprehension of the invention and should not be construed to limit the scope thereof.